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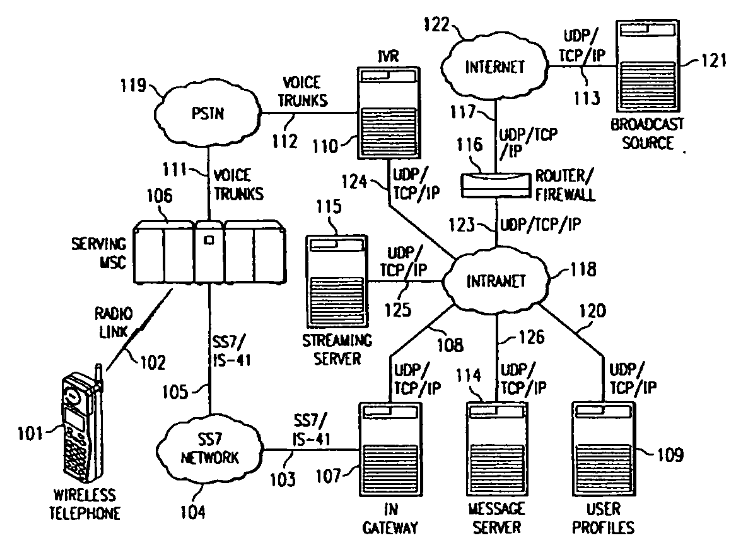
(54) Title: SYSTEM AND METHOD FOR STREAMING INTERNET AUDIO AND VIDEO TO A WIRELESS TELEPHONE



(57) Abstract: The disclosed invention provides a system and method for streaming audio and video signals from the Internet to a wireless device. A telephone network receives a call setup request from a subscriber. The dialed digits in the call setup request are matched to a desired audio or video source, such as an Internet address or a network server. The telephone network opens a channel to the audio or video source and begins receiving streaming audio or video packets. At the same time, the telephone network opens a communication link to the user's wireless or wireline telephone. Streaming audio or video packets are converted to the appropriate format for the user's telephone, such as PCM signals. The converted audio or video stream is then sent to the user's telephone to be played for the user.

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SYSTEM AND METHOD FOR STREAMING INTERNET AUDIO AND VIDEO TO A WIRELESS TELEPHONE

TECHNICAL FIELD

The present invention relates to the provision of Internet data to wireless telephones and, more particularly, to a system for converting Internet streaming media, such as audio and video packets, to a format that can be used by a wireless telephone.

BACKGROUND

Internet users can access many different types of multimedia files and sources that allow the user to hear, play or view sounds, music, animation and video. Typically, these files are accessed through an Internet connection from the user's Personal Computer (PC). Well known sound file formats include the WAV format (files ending in the ".wav" extension) and the MP3 format (files ending in the ".mp3" extension). WAV and MP3 sound files can be downloaded from the Internet and then played by the user on a PC or other device. Other sound files, such as those created by RealAudio® software, allow the user to listen to the sounds while the file is downloading. This process is known as audio streaming.

Users can also obtain video files via the Internet. Common video formats include the MPEG format (files ending in .mpg) and Macintosh QuickTime. Windows ".avi" files can be used for animation. Like the WAV and MP3 files, it is typical for these video and animation files to be completely downloaded by the user and then played. Other files use special software so that the user can watch the video while it is downloading. This capability is known as video streaming.

Streaming audio and/or video techniques provide a method of sending the audio or video data over the Internet as a continuous, compressed stream that is played for the user on the fly as it is received. Streaming allows the user to listen to or view received information almost as soon as it is received, as compared to playing downloaded audio or video files which may have a delay of several minutes while the entire file is transferred. Although no universally-supported streaming standard has been defined, a widely used standard for audio streaming is RealAudio®.

Streaming technology not only allows users to play prerecorded audio and video files, such as books, music or videos that have been digitized and stored, but also allows users to receive live audio and video in realtime. For example, radio stations and television stations can stream their broadcasts over the Internet or any other computer network. Any other source of live or recorded audio or video can also be streamed to send users information or entertainment via the Internet. Users can listen to or watch these broadcasts using, for example, RealPlayer® software or Windows® Media Player.

When a user selects a sound clip on the Internet, such as by clicking on a Web page link on the PC's Web browser, the link typically does not lead directly to a sound file. Instead, the Web browser contacts the Web server, which sends a metafile to the browser. The metafile contains the true location (Internet/network address or Universal Resource Locator (URL)) of a server that hosts the selected sound file. The metafile also contains instructions directing the Web browser to launch an audio player. Typically, the audio player is a software plug-in that is designed to work with the Web browser. Once the audio player is launched, it contacts the URL contained in the metafile. This URL may not be on the original Web server, but instead may be for a separate server that is designed to deliver streaming audio or video clips, such as a RealPlayer® server that delivers RealPlayer® sound clips.

The audio player and the streaming audio server exchange information, such as the connection speed between the user and the Internet. Higher speed or wider bandwidth connections allow the audio server to send more data, which corresponds to a higher-quality sound file and, thus, higher-quality sound played by the PC. For lower connection rates, the sound quality may be diminished. The sound file, which has been compressed and encoded, is sent in IP packets using the User Datagram Protocol (UDP) instead of the more widely used Transmission Control Protocol (TCP). UDP does not resend packets if there are problems in the transmission to the user. If missing or corrupted packets were resent, the sound player would be continuously interrupted and could not play the sound clip properly. On the other hand, the Real-time Transport Protocol (RTP) timestamps the packets so that missing packets can be detected and compensated for by the receiving audio or video codec.

The received packets are stored in a buffer on the user's PC. When the buffer is full, the audio player begins playing the sound clip. Usually, the audio player allows the user to move to different places in the sound clip to replay the sound or to jump ahead to a later portion of the clip. When the user changes places in the sound clip, the audio player contacts the streaming audio server and tells the server to start sending packets from the new position.

Increasingly, radio stations are providing live audio streaming over the Internet. These radio station include traditional radio stations that broadcast over the airwaves and

newer "Internet radio stations" that offer Internet-only broadcasts. Music, news or other sounds are played or broadcast by the radio station. The broadcast information is then converted to a digitized format that can be processed by computers and sent over the Internet. These formats include the RealPlayer® and Windows® Media Player formats. An Internet server hosts the live broadcast in the converted, digital format.

When a user wants to listen to a radio broadcast via the Internet, they launch a client software radio player. From the radio player, the user either selects a link or enters a URL for the desired Internet radio broadcast. When the link is selected or clicked, the radio player software contacts the hosting server, which then sends the broadcast to the user's PC in a steady stream similar to the transmission of an audio clip. The radio broadcast is then played on the user's PC. Generally, the higher the connection speed or the wider the bandwidth, the higher the quality of the broadcast.

Video streaming is accomplished in basically the same manner as audio streaming. Both live video and previously stored video files can be streamed over the Internet. The original video image is captured, digitized and compressed into an appropriate format. To reduce the amount of video data, interframe compression may be used so that only those pixels that change from one frame to another are transmitted. The video information is stored or hosted by a video server on the Internet. Users may view video clips by using a software video player. Packets of video data are sent to the user's PC over the Internet connection using the UDP protocol. As the IP packets are received, they are buffered and then played for the user. Usually, the video information is discarded after it is played rather than being saved on the PC.

A special Internet high-speed backbone, called the Multicast Backbone (MBone), that is capable of sending very large amounts of information can be used to send video transmissions to a number of users simultaneously. Transmission via the MBone uses an IP Multicast protocol instead of the TCP protocol. The Multicast protocol allows the packets to be sent to many users by putting information about multiple Internet destinations within the packets. Initially, the transmission goes out as one file. As the files traverse the Internet, copies of the files are made when appropriate to ensure that the files reach each user. For example, a live video stream may be sent via the MBone to ten users that share the same Internet Service Provider (ISP). A single copy (instead of ten

copies) of the file is sent from the video source server to the ISP's server. The ISP server then copies the files and sends a file to each of the ten requesting users.

Videoconferencing is also possible via the Internet. Two users may exchange video information directly from one Internet address to another. Alternatively, two or more users may conduct an Internet videoconference by using a "reflector" server that sends the video signals from each participant to all of the other participants. Whiteboard applications, which allow two or more participants to work live on the same computer display from different locations, are also possible with video streaming.

The prior art methods of streaming audio and video information require the user to access a PC or some other computer for connection to the Internet. Streaming audio and video information has not been available to users via a wireless or wireline telephone. There are systems that allow users to receive text information via a wireless telephone. For example, the Short Message Service (SMS) of a wireless system can be used to provide information on-demand to a user. Such a system is disclosed in pending patent applications serial number 09/317,476, filed May 24, 1999, and serial number 09/344,407, filed June 24, 1999, both entitled System and Method for Providing Subscriber-Initiated Information Over the Short Message Service (SMS) or Via a Microbrowser and commonly assigned with the present application, the disclosures of which are hereby incorporated by reference herein.

The 09/317,476 and 09/344,407 applications teach a system in which users configure an on-demand profile so that desired information is sent to a wireless device in response to a pre-identified telephone number or code, or in response to an SMS origination message. This on-demand information may be obtained from an Internet server or database, or from some other source. In the referenced applications, the requested information is sent to the wireless device as an SMS text message. This text is then displayed on the wireless device.

However, no system or method exists for streaming requested audio or video broadcasts to a wireless device from an Internet source. Accordingly, it is an object of the present invention to provide audio and video data streams to a wireless or wireline telephone.

It is another object of the invention to provide streaming audio and video that may be either recorded or live. The source of the audio or video stream may be an Internet web server or another node on a computer network.

SUMMARY OF THE INVENTION

These and other objects, features and technical advantages are achieved by a system and method which provides an interface between streaming audio and video Internet servers and a wireless device. Broadcasters and others can provide real-time or on-demand audio and video information via the Internet. The present invention provides a link between such audio or video streams and user telephones.

In one embodiment of the invention, a wireless telephone user configures a user profile to identify a number of preferred services. The user profile associates these preferred services with certain dialed digit strings, feature codes or telephone numbers. When the user initiates a call, the wireless network analyzes the dialed digits, telephone number or feature code to determine if it is associated with a preselected service in the user's profile. When the dialed digits correspond to audio or video streams available on the Internet, the wireless network determines the address or URL for the server that provides the audio or video stream. A connection is established to the streaming server and the wireless network begins receiving audio or video packets via the Internet.

At essentially the same time a connection is established in the normal manner between the wireless network and the user's telephone. The call connection setup messages include information to link the wireless call to the received audio or video stream. An interface device converts the streaming audio or video packets in to the appropriate format that can be sent by the wireless network to the telephone. For example, the streaming audio and video packets may be converted to a PCM format and the signals can be processed and sent to the wireless telephone as if they were received from the public telephone network.

Instead of being part of a wireless network, the present invention may also be part of a wireline network, such as a private network in which calls are routed by a Private Branch Exchange (PBX). The dialed digits or feature codes used in the invention to initiate the streaming audio or video service may be unique for each user, such as those defined in a user profile, and/or they may be generic numbers or codes that are available for use by an subscriber to the system.

In another embodiment, the present invention may be a stand alone system that can receive calls from any subscriber via the PSTN. The stand alone system determines

what service is being requested by the user in any of a number of ways. For example, the system may maintain unique user profiles in a database and the system could provide the user-selected services after the caller was identified. The callers could be identified by Automatic Number Identification (ANI) information for incoming calls. Each subscriber could be assigned a unique dial in number and Dialed Number Information Service (DNIS) could be used to identify the caller. Alternatively, each service could be assigned a unique telephone number and DNIS information could be used to connect the caller to the service that corresponds to the destination number. Other embodiments are also possible, including the use of a interactive menu that prompts the user to select services.

In any case, once the desired services has been identified, the system opens a link to the appropriate host for the audio or video stream. In some cases this will involve establishing a link to a server via the Internet. The service may be identified by a URL or other address and the system establishes a connection using well known procedures for Internet communications. In other cases, the system may communicate with the host device via a Wide Area Network (WAN)/Local Area Network (LAN), a proprietary communications link or some other connection. Once the appropriate connection has been established to the host device, the system begins receiving data packets carrying the audio or video information.

As the streaming audio or video information is received, it is converted to whatever format is required by the connection to the user. This may be a connection via a local switch or PBX, the PSTN or a wireless network. The converted signals are then sent to the user.

In alternative embodiments, the user profiles may include time-based triggers or event triggers that are used to initiate a streaming audio or video connection to the user at a preselected time or upon occurrence of a particular event. The time-based triggers could be for unique, one-time programs or they could be for repeating programs. The event based triggers may be triggered by almost any event as long as it can be monitored by a computer system. For example, a voice recognition unit could monitor audio streams and identify preselected words or phrases, such as names of people, companies, countries, or other locations and the like. Other events may include changes in or values

of numerical parameters, such as financial indices, stock quotations, sports scores, weather parameters and the like.

Upon recognizing a preselected event or time, the present invention would initiate a call to the user and simultaneous open a link to the host for the audio or video stream associated with the event or time. For example, if a financial index, such as a market average or a certain stock price, hit a certain level, then the user may be connected to a financial news service audio stream. In another example, many radio stations have an Internet web page with a streaming audio link. If a user enjoys a particular radio broadcast, then the user could configure his profile so that system establishes a connection to the streamed audio for that broadcast at the same time everyday. Thus, the user would always be able to listen to a selected program even if he was traveling. The system opens a link to the streaming audio host and then calls the user on his wireless telephone whereever the user is currently located.

In other embodiments, the system could be used in reverse to convert the user's speech into streaming audio packets. The user's speech could then treated like any other streaming audio source. The user's remarks could be recorded and stored on an Internet server for later access by others or the speech could be streamed in real time to other parties that receive the words via a PC-based Internet connection. A number of such users could be connected in such a system so that a teleconference could be conducted with some parties accessing the event by wireless or wireline telephone and others accessing the system via a more direct Internet connection, such as a PC. A teleconference type setup could be used for a business meeting or for a classroom or instructional setting.

In wireless networks that are capable of sending large volumes of data to a wireless telephone, the present system may be used without requiring a format conversion in the wireless network. Instead, streaming audio or video packets may be routed from the Internet host directly to the user's wireless telephone. The wireless telephone may then have the required hardware and/or software to convert the streaming audio or video packets to a useable sound or video signal.

The present invention provides a method for routing an audio or video stream to a user via a wireless or wireline telephone network connection. The method involves a

number steps that may be performed simultaneously or in any order. The telephone network or a device in communication with the telephone network, such as processor or IVR, establishes a connection to an audio or video stream host device via a computer network, such as the Internet. The host device provides audio or video information as a series of data packets over the computer network. The telephone network establishes a voice or data connection to a user's telephone to be used to transmit the streaming audio. The streaming audio or video packets are received from the host device via the computer network.

The audio or video stream is converted to a format that can be used or processed by the user's telephone. The conversion may take place within the telephone network or external to the telephone network, such as within the separate processor or IVR. The converted audio or video signals are sent to the user's telephone. If the telephone network is a wireless network, the audio or video stream may be sent to a wireless handset or other wireless device, and the converting step may be performed in the wireless telephone or device.

The audio and video streams may be received from one or more host devices and the decoded audio and video streams may be combined into a multimedia signal comprising all or parts of the audio and video streams from the one or more host devices.

The requested or desired streaming audio or video may be identified in a call setup message from the user's telephone. The telephone network or separate processor may analyze the dialed digit stream and identifies the requested audio or video stream from the digits in the call setup message. The dialed digits may be compared to one or more user profiles to identify a requested audio or video stream. The requested audio or video stream may be identified by dialed digits or by a feature code in a call setup message. Alternatively, the service may be identified by a menu selection made on a wireless device or wireline telephone or the like.

The user profiles may be maintained in a database with other user profiles. The user profiles comprise information identifying at least one audio or video host device or service. This information is associated with a dialed digit string, feature code, menu selection, or the like, which the user will enter to initiate an audio or video streaming request. Alternatively, the connections to a streaming host device and to the user's

telephone may be initiated without user action upon occurrence of an event specified in said user profiles. The event may be based upon a specific time or upon the value of a monitored parameter.

The format of a received streaming audio signal can have an MP3 or RealAudio® format or any other format that allows transmission of audio information as data packets over a computer network. The format that his sent to the user's telephone may be a WAV format, a PCM format or any format that allows the telephone to play a substantially continuous audio signal to the user.

The system of the present invention provides an Internet media stream to a user's wireless or wireline telephone. The media stream may be an audio stream or a video stream or a combination of both. The system comprises means for receiving an Internet media stream from at least one host device. The receiving means may be a processor or an application running on a PC or server. The receiving means is capable of establishing a connection to a media hosting device, such as a server or computer. The connection may be over a computer network, such as the Internet or a WAN/LAN, or over any data connection that allows for the transfer of streaming media data packets.

The system also comprises means for establishing a call connection to the user's telephone. This call connection means may be a component of a wireless or wireline network or a suitable processor in a separate device. The call connection means establishes a call connection in a well known manner between the user's telephone and the system.

The media streaming system also comprises means for converting the received media stream to a format that can be sent to the user's telephone over said call connection. The media stream may be sent over the call connection in a number of ways. For example, the converting means may convert the streaming media packets to an audio signal, such as a Pulse Code Modulation (PCM) signal, that can be sent to the user's telephone in the same manner as a normal voice call connection. Alternatively, the streaming media packets may be formatted for transmission to the user's telephone over the call connection where they are then converted to a useable audio signal for the user. The conversion may be performed by a processor, computer or other device on the

wireless or wireline network. Alternatively, a processor, Application Specific Integrated Circuit (ASIC), or other device in the telephone itself may perform the conversion.

A present invention also provides a method of streaming media signals to a telephone network comprising the steps of receiving streamed media packets from a computer network; routing said received media stream to a hardware sound card; converting the media stream using only the software instance of the sound card to decode the media stream; routing the converted media stream to a telephone network.

The streamed media signals are received from an Internet media server selected by a user, and further comprising the step of sending the converted media stream to the user via the telephone network. The media stream is received via a connection to said Internet media server selected by the user. The user may select the Internet media server by dialing a preselected code that is stored in a user profile. The Internet audio server corresponds to a destination telephone number, feature code, function code or menu selection entered by the user. Alternatively, the system may monitor schedules in one or more user profiles and connect to selected audio host devices when indicated said schedules.

The foregoing has outlined rather broadly the features and technical advantages of the present invention in order that the detailed description of the invention that follows may be better understood. Additional features and advantages of the invention will be described hereinafter which form the subject of the claims of the invention. It should be appreciated by those skilled in the art that the conception and specific embodiment disclosed may be readily utilized as a basis for modifying or designing other structures for carrying out the same purposes of the present invention. It should also be realized by those skilled in the art that such equivalent constructions do not depart from the spirit and scope of the invention as set forth in the appended claims.

BRIEF DESCRIPTION OF THE DRAWINGS

For a more complete understanding of the present invention, and the advantages thereof, reference is now made to the following descriptions taken in conjunction with the accompanying drawings, in which:

FIGURE 1 is a block diagram of a wireless system incorporating an exemplary embodiment the present invention;

FIGURE 2 illustrates some of the signals that may be exchanged in the exemplary system of FIGURE 1;

FIGURE 3 is a block diagram of an alternative embodiment of the present invention;

FIGURE 4 is a flowchart illustrating the steps used in one embodiment of the invention to connect a user to a streaming audio or video source; and

FIGURE 5 is a block diagram of an further embodiment of the audio streaming system of the present invention.

DETAILED DESCRIPTION

The novel features which are believed to be characteristic of the invention, both as to its organization and method of operation, together with further objects and advantages will be better understood from the following description when considered in connection with the accompanying drawings. It is to be expressly understood, however, that the drawings, schematics or block diagrams are given for the purposes of illustration and description only and are not intended as a definition of the limits of the present invention.

The present invention allows a wireless device to access audio and video information on a computer network, such as the Internet. The wireless network opens a first link to the location or source of the desired audio or video information. Streaming audio or video packets are received over the first link. A second link, such as a voice connection, is opened between the wireless device and the wireless network. The wireless system then converts the streamed information to the proper format for transmission to the wireless telephone.

In the present invention, users can configure a profile identifying various sources of streaming audio or video. These streaming sources may be correlated to dialed digits, function code, speed dial number or any other input for the wireless device. When the user enters the appropriate digits, code or the like, the system references the user profile to determine what service is being requested. The wireless system then creates a link between the wireless device and the audio or video stream, and routes the requested information to the wireless device.

A user may request text information from an Internet source and have the information sent to a wireless telephone via an SMS message. Such a system is disclosed in pending applications 09/317,476 and 09/344,407 referenced above. When audio or video information is requested by the wireless user in accordance with the present invention, the information is provided as an audio or video stream instead of as text in an electronic mail or SMS message. The source of the requested information may be any Internet audio or video server or any such hosting server or processor that can be accessed by a network such as a LAN or WAN.

User selection of the Internet audio and video streaming service is similar to the operation of the on-demand text information system that is disclosed in the referenced

applications. A request is keyed into the user's wireless telephone, a gateway server identifies the dialed digits as an Internet audio or video stream request and sends the request to an audio/video server. The audio/video streaming server opens a link to the requested audio or video source via the Internet or some other network connection. At the same time the network completes a voice call connection to the requesting wireless telephone. The routing number associated with the wireless voice connection matches a routing number for the streaming audio/video link. The network connects the user to the streaming audio or video source by converting the streaming audio signal format to a voice or audio signal, such as a Pulse Code Modulation (PCM) signal, that can be routed to the wireless telephone.

Figure 1 is a block diagram of the components of an exemplary system for streaming audio and/or video information to a wireless device. Figure 2 illustrates some of the messages that may be exchanged between the components in Figure 1.

User personal profiles are stored on server 109. These profiles include a user defined association between certain feature codes, dialed digit strings, or function buttons and various sources of text, audio or video information. These sources may include hosting Internet servers. The information sources are identified by a network address or other location for the Internet server or host. A system and method for configuring personal user profiles is disclosed in pending application serial number 08/996,524, entitled System and Method for Controlling Personal Information and Information Delivery to and from a Telecommunications Device, filed December 23, 1997, the disclosure of which is hereby incorporated by reference herein.

Users may access the Server 109 database via a local or remote PC (not shown) or via wireless telephone 101 to configure the profiles. The user enters a particular dialed digit string and an associated audio channel or video source into the database. The audio channel or video source may be identified by a URL or some other Internet or network address. For example, a particular user may configure his profile so that the digits string "#25" is associated with a sports information source. Thereafter, whenever the user desires to hear the latest sports scores, he only has to dial #25 on mobile phone 101. The wireless system uses the dialed digits to identify the address for the desired service.

The dialed digits "#25" are sent from Wireless Telephone 101 to Mobile Switching Center (MSC) 106 over air interface 102 in the normal manner for a call setup. These dialed digits are shown as message 201 in Figure 2. Upon receiving the dialed digits, MSC 106 sends an OriginationRequest trigger (202) to Gateway 107, which may be configured to act as a Signaling Control Point (SCP). Gateway 107 extracts the Mobile Identification Number (MIN) and the dialed digits from OriginationRequest message (202). Using this information, Gateway 107 queries Profile Server 109 via Message Server 114 to validate the user and to determine what service is being requested (messages 203 and 204). Gateway 107, Message Server 114 and Profile Server 109 exchange messages via wireless network Intranet 118.

Profile Server 109 compares the MIN against the user profile database and verifies that the user is a valid subscriber. Preferably, each MIN is associated with a single user profile. However, more than one user could share the same Wireless Telephone 101. Alternatively, universal or default profiles could be stored in Server 109 for use by all subscribers. Profile Server 109 accesses the user's profile to determine which service corresponds to the dialed digits "#25." In the present example, the user has configured these digits to refer to an address that provides audio streaming of sports scores. Therefore, Profile Server 109 returns the user confirmation and the appropriate sports scores server address to Message Server 114 via message 205.

After receiving message 205, Message Server 114 sends message 206 to Interactive Voice Response (IVR) unit 110 to request an idle telephone number. Message 206 also provides the user's MIN and the address for the channel containing the requested audio feed. IVR 110 sends the Interactive Voice Response unit Directory Number (IVRDN) (message 207) to Message Server 114. The IVRDN and the valid user confirmation are then forwarded to Gateway 107 in message 208.

Gateway 107 returns OriginationRequest return result message 210 to MSC 106 using the IVRDN routing number as the Temporary Local Directory Number (TLDN). MSC 106 uses this routing number to completes a call to IVR 110 in the normal manner. In parallel, IVR 110 assigns the IVRDN to the link to the requested audio channel (209) and awaits streaming setup message 215 from Streaming Client 115.

IVR 110 scans lines 112 for incoming calls. New calls (212) are answered and processed to extract the called number information, which is then compared against the reserved IVRDNs. The IVRDNs are matched to link the caller and the requested service. Alternatively, the IVRDN may be common to all calls and the IVR scans the incoming calls for Calling Line ID (CLID) or Calling Number. The CLID parameter will be equal to the MIN provided by the message server in the setup request message and can be used as the key to match the incoming call to the service. A IVRDN or CLID match confirms the required service for the voice call. IVR 110 may play a recording to the incoming call, such as "Please wait while we connect your service" while completing the link. Upon receiving Stream Setup message 215 from Streaming Client 115, IVR 110 proceeds to build the audio stream requested in the service.

IVR 110 connects the incoming call (213) and makes the Internet audio stream (214) available to the caller by converting from the Internet formats, such as MP3 or RealAudio®, to digital PCM signals or any appropriate format for voice signals in the telephone system. The user then listens to the Internet audio stream for as long as desired and disconnects when he is done listening.

When the user disconnects, IVR 110 releases the Internet audio streaming resource and ends the call connection to Wireless Telephone 101. A billing application may be used to record the length of connection. Alternatively, since the call made by mobile 101 is a normal voice call, it may be included in existing billing arrangements.

In an alternate embodiment, the present invention can be configured so that the user's profile defines a time when wireless telephone 101 should be connected to a particular Internet audio or video stream. A time-based trigger in the user profile initiates a setup request message (206) to IVR 110. Setup message 206 may be an out-dial request to the MIN for wireless telephone 101. In this embodiment, IVR 110 does not return an IVRDN, but instead dials out to wireless telephone 101. Additionally, Message Server 114 issues a Get Channel message (209), whereupon Streaming Client 115 sets up the desired audio or video stream to IVR 110. Upon connection to wireless device 101, IVR 110 connects the user to the correct audio channel.

The user-initiated and the out-dialing embodiments of the invention are not limited to use with wireless telephones. The system is equally applicable to a wireline

telephone. Furthermore, wireless telephone 101 can be any analog, digital, or PCS phone without restrictions as to standard or format. Currently wireless systems employ well-known analog and digital Frequency Division Multiple Access (FDMA), Time Division Multiple Access (TDMA), and Code Division Multiple Access (CDMA) techniques in systems complying with the Global System for Mobile Communications (GSM), Interim Standard (IS) 54/136, IS-95, Advanced Mobile Phone System (AMPS), EIA-553, Pacific Digital Cellular (PDC) Standards and wireless telephony standards. The present invention may be used with any of these systems or with any third generation or future developed wireless systems.

Figure 3 illustrates another embodiment of the present invention. The system of Figure 3 is not limited to wireless networks, but instead illustrates one embodiment in which the PSTN can be linked to the Internet to provide streaming audio. Calls arrive at IVR 306 via PSTN 307. IVR 306 answers the calls and holds them using methods known to those skilled in the art. IVR 306 maybe a standalone system or under the direction of other computer programs that allow the incoming calls to be associated with specific service requests. Wireless telephones can be connected to data applications, as taught in U.S. Patent No. 5,978,672, entitled Mobility Extended Telephone Application Programming Interface and Method of Use, issued November 2, 1999, the disclosure of which is hereby incorporated by reference herein. Using this technique, signals from the wireless network can be sent to database 308 to retrieve user profile data and to resolve which audio stream has been requested.

When a call arrives at IVR 306, the call is held by IVR 306 until application program 304 arranges through API 302 to make the requested audio stream available. The received audio stream is sent to sound card 303. However, application program 304 uses only the software instance of the sound card to decode the received audio stream. The decoded audio stream is directed to IVR 306. This decoding process allows Internet streaming audio packets, which are normally MP3 or RealAudio® format, to be converted to the WAV format or other any other format required by the telephone interface boards in IVR 306.

NetHub software 305 is one example of available code that is needed to communicate with IVR 306 based on Dialogic cards and VOS software. This

configuration allows application code 204 to control the IVR sound channels and to stream audio from the Internet, thereby allowing the correct audio channel to be connected to the correct voice channel.

The system shown in Figure 3 can be used to connect any incoming connection from PSTN 307 with an Internet audio or video stream. In alternate embodiments, system 301 may also comprise hardware and software for converting signals from the PSTN to a streaming audio or video format. Such a system allows a caller from PSTN 307 to send a live audio message to an Internet address.

IVR 306 may be configured as a switch having a number of assigned telephone numbers. Calls in PSTN 307 that use these telephone numbers as the destination number will be routed to IVR 306. The telephone numbers can be associated with a particular streaming audio service. Thus, when the call is connected to a particular telephone number at IVR 306, system 301 identifies a preselected service for that telephone number and opens a link to that service. The converted audio stream from the link is then provided to the caller. Such a system could be used by either wireline or wireless telephones without the need for personal profiles for each user or telephone. Instead, an audio/video service provider can predesignate the telephone numbers so that callers know what specific service will be provided when they dial that number. IVR 306 could use the Dialed Number Information Service (DNIS) information for the incoming call to identify the called number and the desired service.

Alternatively, user profiles indexed by subscriber telephone numbers may be stored in database 308. IVR 306 could use Automatic Number Identification (ANI) or CLID to determine the calling telephone number. This telephone number is then referenced in database 308 to determine the desired service. Instead of using ANI, IVR 306 might prompt the caller for an account number, personal identification number (PIN) or password that can be used to verify the caller's authorization to use the system, to identify a user profile or to prompt the user to select various services.

Database 308 may hold user profiles that identify certain preselected times or events that cause system 301 to initiate a call to a subscriber. At the preset time, IVR 306 dials the user's telephone number and initiates a connection to the preselected audio

source. When the user answers, IVR 306 may play an introductory announcement and then begin streaming the audio to the user.

System 301 could also comprise voice recognition hardware or software and could monitor audio or video streams, such as news or financial radio broadcasts. Subscribers may identify topics of interest, such as a particular company, stock or financial index. When system 301 recognizes the use of the topic word, such as the mention of a selected company in a news broadcast, IVR 306 dials the subscriber's telephone number and streams the source audio to the subscriber.

The system of Figure 3 may also be used for video conferencing or teleconferencing. IVR 306 receives calls for a particular teleconference. The callers' audio signals are converted to audio packets streamed to a reflector site (not shown) on Internet 318. Other participants may use system 301 or a similar system in a different location to also access the reflector server. Alternatively, some participants may access the reflector via their own Internet connections to receive streamed audio for the teleconference. When system 301 receives audio packets from the reflector site, they are converted to PCM or other formats as described above and then sent to the caller via PSTN 307.

Figure 4 is a flowchart that illustrates the steps performed in a wireless network embodiment of the present invention. In step 401, the wireless network receives dialed digits from a wireless telephone. The wireless network, or a third party system, maintains a database of user profiles that allow users to predetermine which services are associated with certain dialed digit strings. In step 402, the wireless network queries the user profile database to identify which service has been requested by the dialed digits. Once the requested service has been identified, such as by a URL or network address, the wireless network opens a link to the server hosting the service. The wireless network also establishes a link to the wireless telephone in step 404. The streaming audio from the Internet source is converted to the required voice format, such as PCM, in step 405. This converted audio is then sent as voice information to the wireless telephone in step 406.

Figure 5 is an alternative embodiment of a system for implementing the present invention. As described above, a user initiates the streaming audio or video by dialing a preselected digit string on wireless device 501. For example, the dialed digit string

"##49" may be used to access a radio broadcast for a particular sporting event. Wireless network 502 receives the dialed digits and recognizes that the digit string does not represent a telephone number. Wireless network 502 routes the dialed digits to a software application that recognizes the dialed digits as a streaming audio request and converts the "##49" digit string to a telephone number that is associated with the requested service.

The telephone number is sent to switch 503, which may be part of the wireless network 502, such as an MSC, or part of an external network, such as a PBX. Using the telephone number associated with the dialed digits, switch 503 completes a call to server 505 having one or more interface cards 504, such as Dialogic® cards. The incoming call is processed by an IVR module 506 on Server 505. IVR module 506 has a first process 507 that validates the user status. Process 507 verifies that the user is authorized to access the audio streaming service. The validation process may reference a database of current valid users (not shown). A second process 508 on server 505 specifies the URL and interface card channel that are to be used for this streaming connection. Process 508 may access to a database containing user profiles. The profiles may be used to correlate the dialed digits to the URL of the requested service. These profiles may be specific to individual users or generic, preselected digit strings that are useable by any subscriber.

Once the URL has been specified for the current connection, audio client 509 can set up the streaming connection between server 505 and Internet 511. A process 510 on audio client 509 receives the URL for the requested service. Using this URL received from IVR 506, the process 510 establishes a connection to the requested audio source via Internet 511 and begins receiving streaming audio packets. The audio data packets are passed to rendering plug-in 512, which buffers the data into memory, decodes the data and sends the data to an audio services interface. This can be achieved using the RealSystem™ G2 SDK. Rendering plug-in 512 converts the streaming audio to a format for media streaming terminal 513.

IVR 506 specifies a channel on interface card 504 of server 505 that is to receive the streaming audio signals. These signals are provided from Internet 511 through audio client 509 and media stream terminal 513 to interface card 504, which routes the streaming audio signals as a continuous audio signal to the user.

In this embodiment, the default audio device for an interface card or IVR is replaced with a customized audio device. For example, if Dialogic card is used, the customized audio device sends data packets to a specific channel on the Dialogic card. The channel is specified by the IVR. The Microsoft TAPI 3.0 can be used to define a media stream terminal to achieve this configuration.

It will be understood that wireless telephone may comprise software and hardware that can receive and display video information. In such a case, the present invention may stream video signals to the user's wireless telephone. In other cases, when the wireless telephone cannot display video, an audio portion of a streaming video signal may be stripped from the streaming signal and sent to the user without the video components. Although the terms audio, video and media are used in various examples throughout the description, it will understood that these terms are interchangeable and that they refer to various types of aural, visual, text or other media that may be streamed to a user. The source of such media may be from a stored source, such as a computer memory, data file or database, or from a live source in real-time or near real-time.

In other applications, such as in wireless networks having enhanced data rates, the wireless network may be capable of sending the streaming audio packets directly to the wireless device. For example, next generation wireless systems may introduce Enhanced Data Rates for GSM and TDMA/136 Evolution (EDGE). These systems will have the capability to provide high data rate services, including high bit-rate, circuit-switched modes and packet services, such as High-Speed, Circuit-Switched Data (HSCSD) and General Packet Radio Service (GPRS). Wireless devices operating in such systems may receive packet data directly from the wireless network a rate sufficient to provide streaming audio and video services. These wireless devices may include hardware and software for converting streaming audio or video files to PCM signals or to audible signals for the user, thereby eliminating or reducing the need for such a conversion in the wireless network.

Although the present invention and its advantages have been described in detail, it should be understood that various changes, substitutions and alterations can be made herein without departing from the spirit and scope of the invention as defined by the appended claims. Moreover, the scope of the present application is not intended to be

limited to the particular embodiments of the process, machine, manufacture, composition of matter, means, methods and steps described in the specification. As one of ordinary skill in the art will readily appreciate from the disclosure of the present invention, processes, machines, manufacture, compositions of matter, means, methods, or steps, presently existing or later to be developed that perform substantially the same function or achieve substantially the same result as the corresponding embodiments described herein may be utilized according to the present invention. Accordingly, the appended claims are intended to include within their scope such processes, machines, manufacture, compositions of matter, means, methods, or steps.

WHAT IS CLAIMED IS:

We claim

1. A method of providing a media stream to a user via a telephone network connection, comprising the steps of:

establishing a connection to a media stream host device, wherein said host device provides streaming media information as a series of data packets to sent over a computer network;

establishing a connection to a user's telephone;

receiving a media data stream from said host device; and

converting said media stream to a format used by the user's telephone.

2. The method of claim 1 further comprising the steps of:

sending the converted media signal to the user's telephone.

3. The method of claim 1 further comprising the step of:

sending the received media stream to a wireless handset; and

wherein said converting step is performed in the wireless telephone.

4. The method of claim 1 wherein two or more media streams are received from one or more host devices; and further comprising the step of:

combining decoded media streams into a multimedia signal.

5. The method of claim 1 further comprising the steps of:

receiving a call setup message from said user's telephone; and

identifying a requested media stream from said call setup message.

6. The method of claim 5 further comprising the steps of:

comparing data from said call setup message to a user profile to identify said requested media stream.

7. The method of claim 6 wherein said data from said call setup message is a group of dialed digits.

8. The method of claim 1 further comprising the steps of:

providing a database of user profiles, said profiles comprising information identifying at least one streaming media host device.

9. The method of claim 8 further comprising the steps of:

initiating said connections to said host device and to said user's telephone upon occurrence of an event specified in said user profiles.

10. The method of claim 8 further comprising the steps of:

initiating said connections to said host device and to said user's telephone at a time specified in said user profiles.

11. The method of claim 1 wherein a received streaming media signal has an MP3 or RealAudio® format.

12. The method of claim 1 wherein said format used by the user's telephone is a WAV format.

13. The method of claim 1 wherein said media stream is a stream of audio data packets.

14. The method of claim 1 wherein said media stream is a stream of video data packets.

15. A system for providing a media stream to a user's telephone, comprising:
means for receiving a media stream from at least one host device;
means for establishing a call connection to the user's telephone; and
means for converting said received media stream to a format that can be sent to said user's telephone over said call connection.

16. The system of claim 15 further comprising:
means for receiving a call setup message from the user's telephone; and
means for identifying a requested media stream from information in the call setup message.

17. The system of claim 16 wherein said information in the call setup message is dialed digits.

18. The system of claim 16 further comprising:
a database of user profiles, said profiles including information that identifies said at least one host device.

19. The system of claim 18 wherein said information that identifies said at least one host device comprises an Internet address for said host device.

20. The system of claim 18 further comprising:
means for comparing said call setup message information to said user profiles to identify a requested media stream.

21. The system of claim 15 wherein said received media stream is a stream of audio data having an MP3 or RealAudio® format.

22. The system of claim 15 wherein signals sent to said user's telephone have a WAV format.

23. The system of claim 15 wherein said received media stream is a stream of video data.

24. A method of providing streaming audio signals to a telephone network comprising the steps of:

receiving streamed audio packets from a computer network;

routing said received audio stream to a hardware sound card;

converting the audio stream using the software instance of the sound card to decode the audio stream; and

routing the converted audio stream to a telephone network.

25. The method of claim 24 wherein said streamed audio signals are received from an Internet audio server selected by a user, and further comprising the step of:

sending the converted audio stream to the user via said telephone network.

26. A method of providing streaming media signals to a communications network comprising the steps of:

specifying an address for a streaming media source;

specifying a channel in said communications network that is to receive said steaming media signals;

establishing a connection to a streaming media source associated with said address;

receiving streaming media packets from said source;

converting said streaming media packets to a format usable on said communications network; and

routing said converted streaming packets to said channel in said communications network.

27. The method of claim 26 wherein said channel in said communications network is accessed through a Dialogic® card.

28. The method of said 26 wherein said communications network is a wireless telephone network.

29. The method of claim 26 wherein said address for said streaming media source corresponds to a preselected code dialed by the user.

30. The method of claim 26 further comprising the steps of:
monitoring schedules in one or more user profiles; and
connecting to selected media host devices when indicated in said schedules.

31. The method of claim 26 wherein said streaming media signals are streaming audio signals.

32. A system for providing access to Internet audio sources comprising:
means for connecting to an Internet audio host device;
means for receiving streamed audio packets from said Internet audio host;
means for sending said audio packets to a wireless telephone; and
means, at said wireless handset, for decoding said streamed audio packets into a useable audio signal.

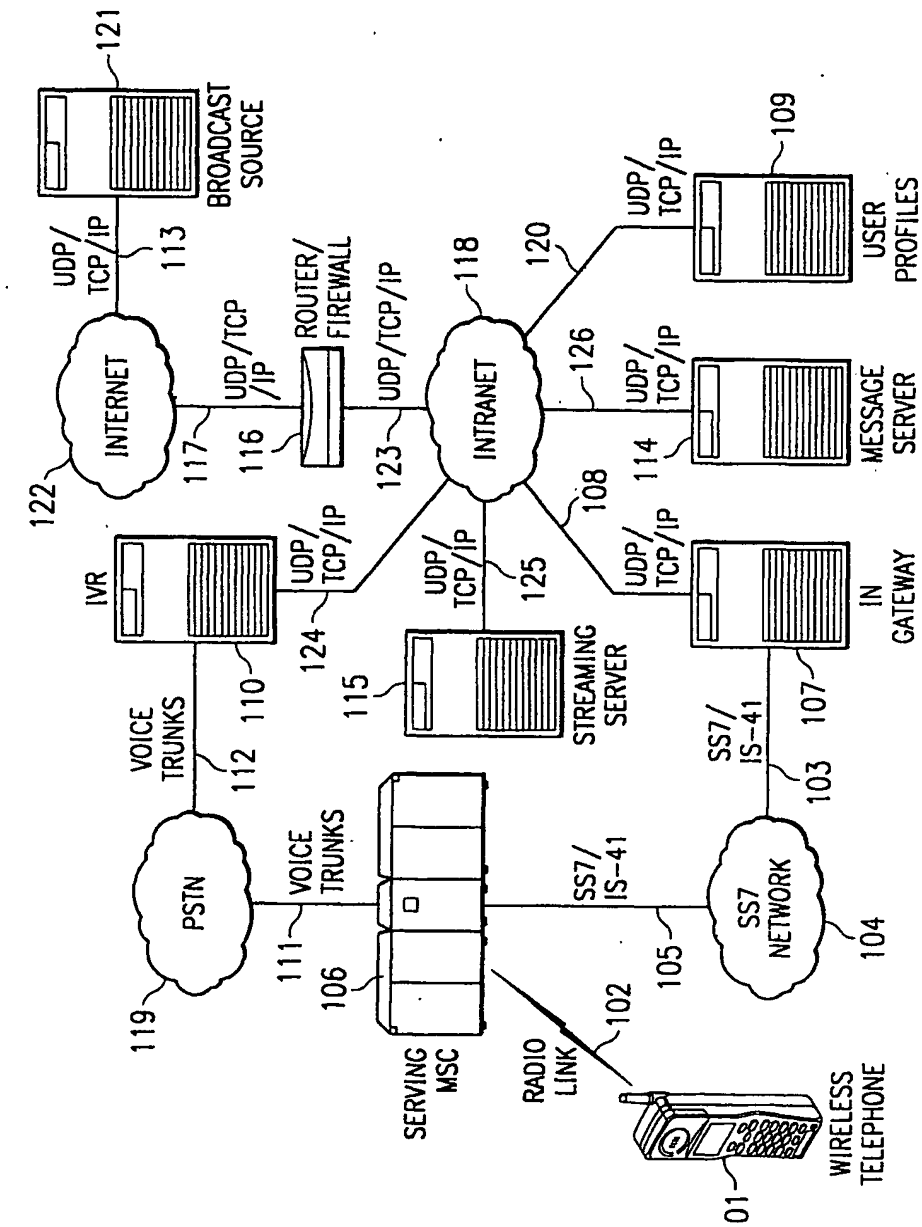
BROADCAST SOURCE
121
UDP/TCP/IP
113
INTERNET
122
117
UDP/TCP/IP
116
ROUTER/FIREWALL
UDP/TCP/IP
123
INTRANET
118
120
UDP/TCP/IP
USER PROFILES
109
126
UDP/TCP/IP
MESSAGE SERVER
114
108
UDP/TCP/IP
IN GATEWAY
107
IVR
110
UDP/TCP/IP
124
UDP/TCP/IP
125
STREAMING SERVER
115
VOICE TRUNKS
112
PSTN
119
VOICE TRUNKS
111
SERVING MSC
106
SS7/IS-41
105
SS7/IS-41
103
SS7 NETWORK
104
RADIO LINK
102
WIRELESS TELEPHONE
101

FIG. 1

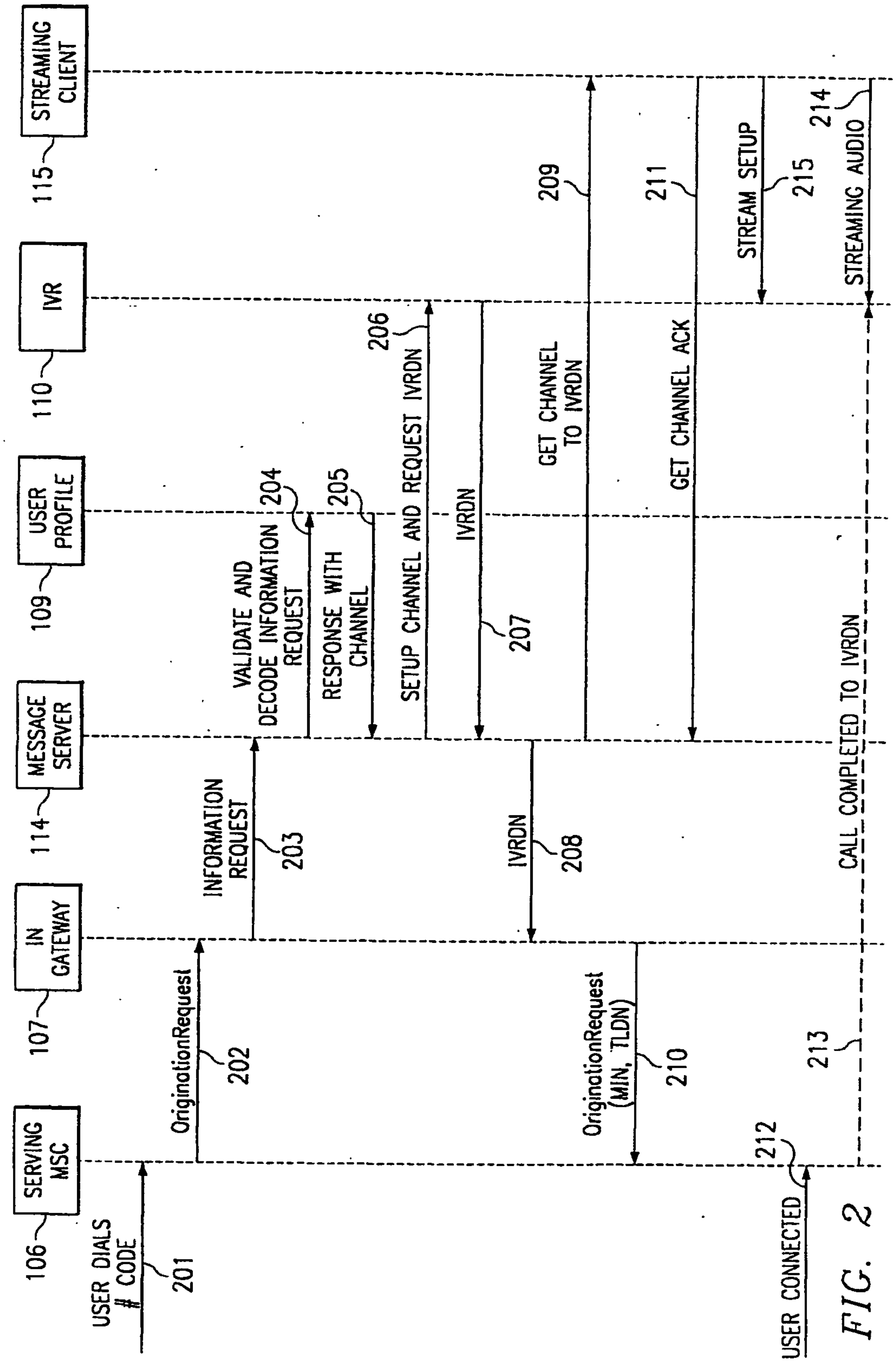
SERVING MSC
106
IN GATEWAY
107
MESSAGE SERVER
114
USER PROFILE
109
IVR
110
STREAMING CLIENT
115
USER DIALS # CODE
201
OriginationRequest
202
INFORMATION REQUEST
203
VALIDATE AND DECODE INFORMATION REQUEST
204
RESPONSE WITH CHANNEL
205
SETUP CHANNEL AND REQUEST IVRDN
206
IVRDN
207
IVRDN
208
GET CHANNEL TO IVRDN
209
OriginationRequest (MIN, TLDN)
210
GET CHANNEL ACK
211
USER CONNECTED
212
CALL COMPLETED TO IVRDN
213
STREAMING AUDIO
214
STREAM SETUP
215

FIG. 2

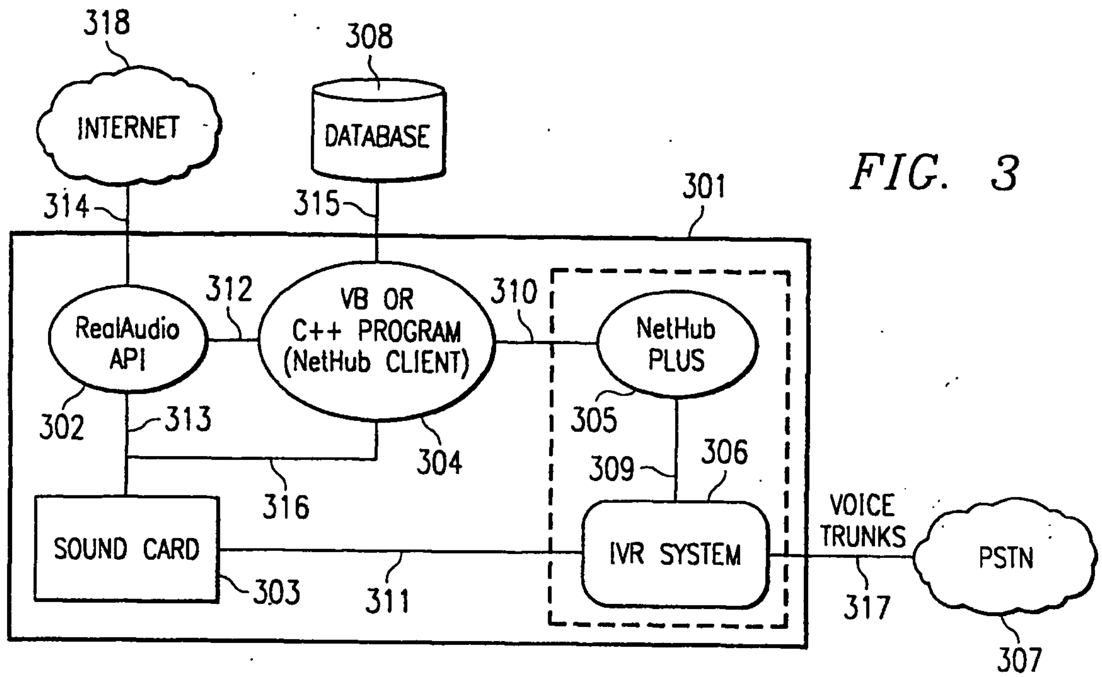
318
INTERNET
308
DATABASE
314
315
301
312
VB OR
C++ PROGRAM
(NetHub CLIENT)
310
RealAudio
API
NetHub
PLUS
302
313
305
304
309
306
316
VOICE
TRUNKS
SOUND CARD
IVR SYSTEM
PSTN
303
311
317
307

FIG. 3

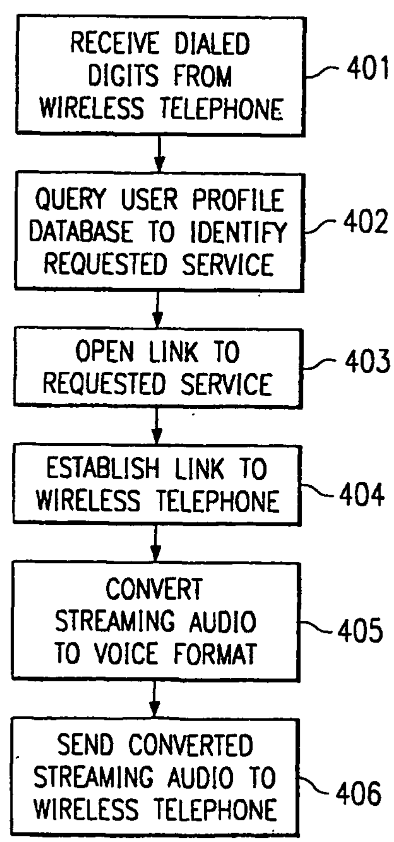
RECEIVE DIALED DIGITS FROM WIRELESS TELEPHONE
401
QUERY USER PROFILE DATABASE TO IDENTIFY REQUESTED SERVICE
402
OPEN LINK TO REQUESTED SERVICE
403
ESTABLISH LINK TO WIRELESS TELEPHONE
404
CONVERT STREAMING AUDIO TO VOICE FORMAT
405
SEND CONVERTED STREAMING AUDIO TO WIRELESS TELEPHONE
406

FIG. 4

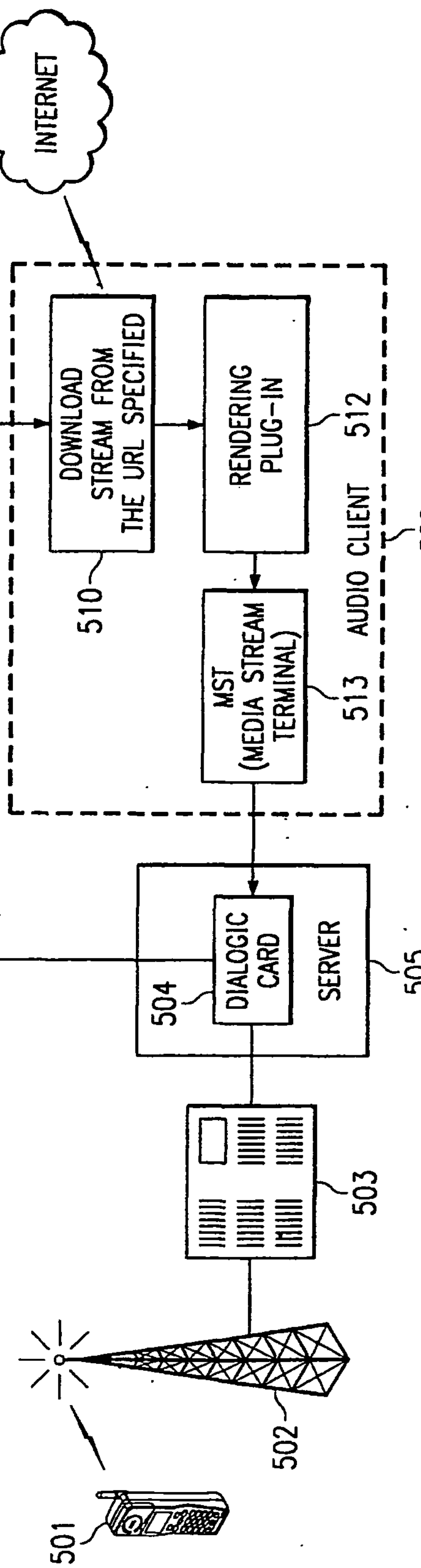
511
INTERNET
506
IVR CENTER
507
VALIDATE USER STATUS
508
SPECIFY THE URL AND DIALOGIC CARD CHANNEL
510
DOWNLOAD STREAM FROM THE URL SPECIFIED
512
RENDERING PLUG-IN
513
MST (MEDIA STREAM TERMINAL)
AUDIO CLIENT
509
504
DIALOGIC CARD
SERVER
505
503
502
501

FIG. 5